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EXAMINER	
SELLERS, DANIEL R	
ART UNIT	**PAPER NUMBER**
2644	

DATE MAILED: 10/29/2004

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)
	09/431,371	SCARPINO ET AL.
	Examiner	Art Unit
	Daniel R. Sellers	2644

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-42* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-42* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *01 November 1999* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date *see attachment*.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

Continuation Sheet (PTOL-326) Application No.

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DETAILED ACTION

Claim Rejections - 35 USC § 102

1. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

2. Claims 1-5, 7, and 8 are rejected under 35 U.S.C. 102(b) as being clearly anticipated by Hausman et al. (Hausman), U.S. Patent 5,262,974.

3. Regarding claim 1, see Hausman

A digital filter comprising a series of digitized time coefficients stored in a memory (Col. 2, lines 58-60), *said time coefficients being mapped to a like number of frequency coefficients, said frequency coefficients being spaced at frequency intervals, having either zero phase angles or linearly spaced phase angles and having amplitudes which are mirrored about a mid frequency to produce periodicity of a time response for said digital filter.* (Col. 4, lines 26-32).

Hausman teaches a linear phase, symmetric digital finite impulse response (FIR) filter. It is inherent in the art of digital filters that the time coefficients are mapped to the same number of frequency coefficients, and that the frequency coefficients are spaced at frequency intervals.

4. Regarding claim 2, the further limitation of claim 1, see Hausman

... said time coefficients are odd in number (Table 1, row 1).

Hausman teaches an odd length filter.

5. Regarding claim 3, the further limitation of claim 1, see Hausman

... the number of said time coefficients is equal to or greater than 5. (Table 1, row 1).

Hausman teaches a filter with 49 coefficients.

6. Regarding claim 4, the further limitation of claim 3, see Hausman

... the number of said time coefficients is one of 7, 9 and 11 (Col. 2, lines 26-32).

Hausman teaches a programmable filter with as many as 49 coefficients.

7. Regarding claim 5, the further limitation of claim 1,

... said time coefficients are defined by inverse discrete Fourier transforms (IDFT) *of said frequency coefficients.*

It is inherent in the art that time coefficients are defined by IDFT's of frequency coefficients. One of ordinary skill in the art chooses a frequency response first and then computes the corresponding coefficients to achieve a desired response.

8. Regarding claim 7, the further limitation of claim 1,

... said frequency coefficients are spaced at equal frequency intervals.

It is inherent in the art of digital filters that the frequency coefficients are spaced at equal frequencies.

9. Regarding claim 8, the further limitation of claim 1,

... said time coefficients are integer numbers. (Col. 1, lines 31-33).

Hausman teaches a programmable canonic signed digit (CSD) digital filter. The integer numbers are powers of two.

10. Claims 23, 24, 26, 28-33, and 35 are rejected under 35 U.S.C. 102(b) as being clearly anticipated by Kodra, U.S. Patent 5,566,101.

11. Regarding claim 23, see Kodra

A method of enhancing a series of digital audio samples comprising the steps of: receiving said series of digital audio samples (Fig. 4, unit 110), *and generating a driving signal by convolving said series of samples in real time with a series of stored time coefficients* (Fig. 4, units 132 and 134), *said time coefficients being mapped to a like number of frequency coefficients, said frequency coefficients being spaced at frequency intervals, having either zero phase angles or linearly spaced phase angles and having amplitudes which are mirrored about a mid frequency* (Col. 2, lines 30-32).

Kodra teaches a real-time linear phase, symmetric digital FIR filter with seven coefficients for use in an audio system.

12. Regarding claim 24, the further limitation of claim 23, see Kodra

... further comprising the step of generating an analog audio signal from said driving signal (Col. 3, lines 42-48).

Kodra teaches the use of a digital-to-analog converter (DAC) for generating the analog audio signal.

13. Regarding claim 26, the further limitation of claim 23, see the rejection of claim 23. Kodra teaches the convolution step, and the method as stated in claim 26 is convolution.

14. Regarding claims 28-30, the further limitations of claim 23, see Kodra

... wherein said receiving step comprises the step of reading said series of digital samples from a digital recording medium (Col. 2, lines 58-61).
... said receiving step comprises the step of reading said series of digital samples from a compressed file.
... said receiving step comprises the step of downloading audio sample streams from the Internet.

Kodra teaches that the audio source can be any analog or digital source.

15. Regarding claim 31, see the rejection of claim 23.

Apparatus for enhancing a series of digital audio samples comprising:
a device for receiving said series of digital samples, a digital filter comprising a series of stored time response coefficients, said time response coefficients being mapped to a like number of frequency response coefficients, said frequency response coefficients being spaced at frequency intervals, having phase angles of zero and having amplitudes which are mirrored about a mid frequency, and a microprocessor for generating a driving signal by convolving said sound samples in real time against said time coefficients.

Kodra teaches a real-time system (Col. 2, lines 40-43).

16. Regarding claim 32, the further limitation of claim 31, see the rejection of claims 23 and 31. Kodra teaches a digital signal reader.

17. Regarding claim 33, the further limitation of claim 31, see the above rejections of claims 23, 26, and 31. Kodra teaches the convolution step.

18. Regarding claim 35, the further limitation of claim 31, see the above rejections of claims 23, 24, and 31. Kodra teaches the generation of an analog audio signal.

Claim Rejections - 35 USC § 103

19. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

20. Claim 6 is rejected under 35 U.S.C. 103(a) as being unpatentable over Hausman as applied to claim 1 above, and further in view of Scheerer et al. (Scheerer), U.S. Patent 5,327,137.

... a portion of said frequency coefficients having frequencies within a free band are selected so as to achieve a generally constant oscillation frequency across a center band which is broader than said free band.

Hausman teaches a symmetric digital FIR filter. However, Hausman does not teach a filter that has a constant oscillation frequency across the center band. Scheerer teaches a higher order noise shaper that employs a digital FIR or IIR filter after noise shaping in an analog-to-digital converter (ADC). Scheerer does not specifically teach if the filter is of even or odd length. Scheerer, however, does teach a linear phase equiripple FIR filter (Col. 17, lines 18-30). The equiripple filter has a generally constant oscillation frequency in the passband and the stopband. It would have been obvious to one of ordinary skill in the art to combine the teachings of Hausman and Scheerer to create a linear phase, symmetric digital FIR filter with a constant oscillation frequency across a center band, for the purpose of creating a more precise filter.

21. Claims 9-11 are rejected under 35 U.S.C. 103(a) as being unpatentable over Hausman and Kimura, U.S. Patent 5,172,358.

22. Regarding claim 9, see Kimura

A method of making a digital filter comprising the steps of: establishing a plurality of frequency response coefficients separated at frequency intervals (Col. 4, equation 3), *said frequency response coefficients being graphically characterized by a base point, a series of principal points and a series of mirror points and having either zero phase angles or linearly spaced phase angles, said base point having a frequency of zero, a portion of said principal points being situated at frequencies encompassing the range of human hearing* (Fig. 6) *and a portion of said mirror points having frequencies and amplitudes which mirror said portion of said principal points when viewed relative to a mid frequency higher than said frequencies of said principal points, performing inverse discrete Fourier transformations to map said frequency response coefficients into corresponding time response coefficients, and storing said time response coefficients in a digital memory* (Col. 4, lines 47-49).

Kimura teaches a loudness control circuit for use in audio. Kimura does not teach a zero or linear phase circuit, nor does he specifically teach a base point at a frequency of zero. However, the frequency response of digital filters always includes a frequency coefficient corresponding to a frequency of zero. It is well known in sampling theory that the frequency response of a filter above half of the Nyquist rate is a mirror image of the frequency response of the filter below half of the Nyquist rate. It is also well known in the art to perform an IDFT on a set of frequency coefficients to obtain the time response coefficients. Hausman teaches a linear phase, symmetric digital FIR filter that also stores coefficients in memory. It would have been obvious to one of ordinary skill in the art to combine the teachings of Kimura with the linear phase of Hausman in order to achieve an optimal result for audio reproduction.

23. Regarding claim 10, the further limitation of claim 9, see Kimura

... said portion of said principal points are situated at predetermined frequencies within the range of human hearing and have amplitudes that roughly are inversely corresponding to human hearing sensitivity at said predetermined frequencies (Fig. 6).

Kimura teaches the use of Fletcher-Munson curves. Fletcher-Munson curves are curves that were obtained experimentally by Fletcher and Munson at Bell Labs, and these curves correspond inversely to the sensitivity of human hearing at different sound pressure levels. It would have been obvious to one of ordinary skill in the art to combine this frequency response with the previous combination of Kimura and Hausman for the purpose of better audio reproduction at a given sound pressure level.

24. Regarding claim 11, the further limitation of claim 9,

... said frequency response coefficients are established at uniformly spaced frequency intervals.

It is well known in the art to use uniformly spaced frequency intervals in digital filters.

25. Claim 12 is rejected under 35 U.S.C. 103(a) as being unpatentable over Hausman and Kimura as applied to claim 9 above, and further in view of Suzuki, U.S. Patent 6,298,361.

26. Regarding claim 12, the further limitation of claim 9, see Suzuki

... said establishing and performing steps comprise the following steps:
selecting a plurality of first frequency response coefficients separated at uniformly spaced frequency intervals, said first frequency response coefficients having either zero phase angles or linearly spaced phase angles and each first frequency response coefficient further having an amplitude and a frequency,
arranging said plurality of first frequency response coefficients in order from lowest frequency to highest frequency to define a list of first frequency response coefficients, (Fig. 18b)

performing inverse discrete Fourier transformations to map said plurality of first frequency response coefficients into corresponding first time response coefficients,
discarding a pair of said first time response coefficients which have equal magnitudes and are positioned adjacent to one another in said list, remaining time response coefficients defining second time response coefficients (Col. 15, lines 35-41),
assessing the effect on a frequency response of the digital filter after discarding said pair of said first time response coefficients (Fig. 28a and 28b),
repeating said performing, discarding and assessing steps until a pair of discarded time response coefficients cause a significant change in the frequency response of the digital filter, and adding to remaining time response coefficients said pair of discarded time response coefficients causing a significant change in the frequency response of the digital filter, said added and remaining time response coefficients comprising final time response coefficients.

As previously stated, it is inherent in the art to select frequency response coefficients separated at uniformly spaced intervals, and it is inherent to use the IDFT to map the frequency coefficients to time coefficients. Suzuki teaches a band synthesizing filter bank for real-time processing. He does not teach the use of linear phase filters in the filter bank. Suzuki does not teach the repetition of discarding coefficients, however Suzuki shows two truncated impulse responses derived from the original impulse response. It is inherent that the second truncated response could have been derived from the first truncated response (Fig. 27a and 27b). Suzuki also graphically shows the differences between the two truncated responses (Fig. 28a and 28b). It is inherent that when performing and assessing the truncation of a response, one skilled in the art would recognize when the resulting frequency response fails to meet predefined criteria, and would select an appropriate compromise. The combination of Hausman and Kimura teach the use of linear phase FIR filters, however they do not teach a method of discarding coefficients to reduce the order of the filter. Hausman does teach the use of CSD multiplication, which is performed with shift registers (Col. 1, lines 34-36), which reduces the filter complexity. It would have been obvious to one of ordinary skill in the

art to combine the teachings of Suzuki with the combination of Hausman and Kimura to further reduce the complexity of the filter.

27. Regarding claims 13 and 14, the further limitations of claim on claims 12 and 13 respectively, see the rejection of claims 8 and 12 above.

(13) A method according to claim 12, further comprising the steps of: multiplying each of said final time response coefficients by an integer conversion number to define converted final time response coefficients, said conversion number being sufficiently large to permit discarding any remaining fractional portion without losing substantial final time response coefficient accuracy, and discarding from each of said converted final time response coefficients any remaining fractional portion.

Furthermore, regarding claim 13, it is inherent to convert the floating-point binary number to an integer number by multiplying by a large conversion number. It is well known in the art that quantization effects are to be expected when implementing digital filters, and these quantization effects are analogous to accuracy that may be lost by discarding a fractional part of a large number.

28. Regarding claim 15, the further limitation of claim 12, see the above rejection of claim 12. Suzuki compares two frequency response curves (Fig. 28a and 28b).

29. Claims 16-22 are rejected under 35 U.S.C. 103(a) as being unpatentable over Hausman and Kimura as applied to claim 9 above, and further in view of Bauer et al. (Bauer), U.S. Patent 3,594,506.

30. Regarding claim 16, the further limitation of claim 9, see Bauer

... each of said frequency response coefficients having an amplitude and a frequency and said range of human hearing being within a band of frequencies having a low end and a high end, a portion of said frequency response coefficients having frequencies between a reference frequency and said high end increase in amplitude as per increasing frequencies from said reference frequency toward said high end. (Fig. 3, traces 18 and 53).

Bauer teaches a loudness level indicator, which comprises a network having a transfer function that is inversely proportional to human hearing sensitivity (Col. 3, lines 1-5, and lines 10-13). Trace 18, in figure 3, of Bauer shows an increase in amplitude as frequency increases. Bauer does not teach a linear phase digital filter, which stores coefficients in memory. It would have been obvious to one of ordinary skill in the art to combine the teachings of Bauer with the combination of Hausman and Kimura for the purpose of better audio fidelity.

31. Regarding claim 17, the further limitation of claim 16, see Bauer

... said reference frequency is in the range of from about 501 Hz to about 8018 Hz. (Fig. 3)

One kilohertz is a reference frequency that is common in the art of audio.

32. Regarding claims 18 and 19, the further limitation of claim 16 and 18 respectively, see Bauer

... said frequency response coefficients having frequencies between said reference frequency and said high end increase in amplitude up to a significant amplitude peak at a peak high frequency and decrease in amplitude as per increasing frequencies toward said high end above said peak high frequency. (Fig. 3, trace 18).

And

... said peak high frequency is in the range of from about 1002 Hz to about 20045 Hz. (Fig. 3, trace 18).

Bauer teaches a frequency response that peaks at roughly 10 kHz, and decreases afterward.

33. Regarding claim 20, the further limitation of claim 18, see Bauer

... the amplification of the frequency response coefficient at said peak high frequency is from about 1.3 times to about 6.0 times the amplification of said frequency response coefficient at said reference frequency. (Fig. 3, trace 18).

Bauer teaches a peak in the frequency response that is about 5-10 decibels (dB) larger than the reference frequency of 1 kHz. It is well known that a 3 dB change in amplitude is about equivalent to a doubling of magnitude, and a 6 dB change is about equal to a quadrupling of magnitude.

34. Regarding claim 21, the further limitation of claim 16, see Bauer

... said frequency response coefficients having frequencies between said reference frequency and said high end increase in amplitude up to a significant amplitude peak at a peak high frequency, decrease in amplitude as per increasing frequencies down to a significant amplitude trough at a trough high frequency and increase in amplitude as per increasing frequencies toward said high end. (Fig. 2, trace Robinson-Dadson).

Bauer teaches the contours of various loudness contours with respect to their own contour as shown in Figure 3. The trace of Robinson-Dadson has a peak just slightly above 10 kHz, and shows a trend of reaching a trough in the high end. It is inherent as stated previously that the mirror image is a result of sampling theory and the trace will increase again after this trough to another peak, which will be a mirror of the peak that is roughly at 10 kHz.

35. Regarding claim 22, the further limitation of claim 9, see Bauer

... each of said frequency response coefficients having an amplitude and a frequency and said range of human hearing being within a band of frequencies having a low end and a high end, a portion of said frequency response coefficients having frequencies between a reference frequency and said low end increase in amplitude as per decreasing frequencies from said reference frequency toward said low end. (Fig. 3, trace 53).

Bauer teaches a contour that increases in amplitude as the frequency decreases from a reference frequency of 1 kHz.

36. Claim 25 is rejected under 35 U.S.C. 103(a) as being unpatentable over Kodra as applied to claim 23 above, and further in view of Hausman.

37. Regarding claim 25, the further limitation of claim 23,

... said time coefficients are integer time coefficients.

Kodra teaches a linear phase, symmetric digital FIR filter with 7 coefficients for use in an audio system. He does not teach a FIR filter with integer coefficients. Hausman teaches a linear phase, symmetric digital FIR filter. Hausman further teaches the use of integers that are powers of two in the FIR filter. Hausman does not specifically teach an audio input signal, however Hausman teaches the input of any digital signal (Col. 1, lines 57-59). It would have been obvious to one of ordinary skill in the art to combine the teachings of Kodra with Hausman for the purpose of reducing processing time in an audio system.

38. Claims 36-42 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kodra and Clemow, U.S. Patent 6,208,687.

39. Regarding claim 36, see Kodra,

(36) A filter package having two or more parallel digital filters comprising: a first digital filter comprising a series of digitized first time coefficients stored in a first memory (Fig. 2, items 48, 52, and 64), *said time coefficients being mapped to a like number of first frequency coefficients, said first frequency coefficients having either zero phase angles or linearly spaced phase angles and having amplitudes which are mirrored about a mid frequency* (Col. 2, lines 30-33), *and a second digital filter comprising a series of digitized second time coefficients at least one of which has a value which is different from each of said first time coefficients, said second time coefficients being stored in a second memory and mapped to a like number of second frequency coefficients, said second frequency coefficients having either zero phase angles or linearly spaced phase angles and having amplitudes which are mirrored about a mid frequency.*

Kodra teaches a real-time linear phase, symmetric digital FIR filter with 7 coefficients for use in an audio system. He teaches two memories, and further teaches that the two memories can be a logical partition of one physical memory bank. He does not teach two or more parallel filters. Clemow teaches two digital FIR filters in parallel, and the two filters have coefficients stored in a memory (Col. 2, line 65 – Col. 3, line 2). Clemow does not teach linear phase, symmetric impulse response FIR filters. It would have been obvious to combine the teaching of linear phase filters of Kodra with the parallel processing of Clemow for the purpose of improving audio fidelity.

40. Regarding claim 37, the further limitation of claim 36, see Kodra

... wherein the number of said first time coefficients is equal to or greater than 5 and the number of said second time coefficients is equal to or greater than 5 (Col. 5, line 45 and Fig. 6).

Kodra does not teach two filters, however Clemow does. Clemow does not teach any specific filter order. It would have been obvious to one of ordinary skill in the art to combine the teaching of 5 or more coefficients in Kodra with the previous combination of Kodra and Clemow for the purpose of better audio fidelity.

41. Regarding claim 38, the further limitation of claim 36, see the above rejection of claim 36. Kodra teaches one physical memory split into two distinct parts (Fig. 2, units 48, 50, and 52). Kodra does not teach two parallel filters. Clemow teaches two parallel filters with different coefficients. Clemow does not specifically teach one or two memories for both sets of coefficients. It would have been obvious to one of ordinary skill in the art to combine the teachings of one memory in Kodra with the previous combination for the purpose of cutting manufacturing costs.

42. Regarding claims 39 and 40, see the above rejections of claims 36 and 38. Kodra and Clemow teach filters that receive digital audio inputs.

43. Regarding claim 41, see the above rejections of claims 36 and 39. Clemow further teaches a microprocessor that selects one of the two filters (Col. 3, lines 18-38). It is inherent that convolution takes place when a filter is selected and an input is given.

44. Regarding claim 42, see the above rejections of claims 36, 39, and see Kodra, Figure 1, units 32 and 36. Kodra creates an analog audio signal from the filtered digital drive signal.

45. Claims 27 and 34 are rejected under 35 U.S.C. 103(a) as being unpatentable over Hausman as applied to claim 8, Kodra as applied to claims 26 and 33, and in further view of Iwase et al. (Iwase), U.S. Patent 4,862,403.

46. Regarding claim 27, the further limitation of claim 26, see Iwase

... wherein said step of generating a driving signal further comprises the steps of: dividing said values of Y by a number previously used to convert initial real number time coefficients to said integer time coefficients (Col. 2, lines 28-54); *and discarding any remaining fractional portion of said divided values of Y.*

Iwase teaches a symmetric digital FIR filter with selective scaling factors for its coefficients. Iwase does not teach integer coefficients. Hausman teaches the use of integer coefficients as show in the rejection of claim 8. Hausman does not teach a dividing step. Kodra teaches the step of generating a drive signal, but also does not teach integer coefficients. It is inherent in scaling coefficients in a digital register that some overflow is expected and will result in discarding of fractional portions. Therefore it would have been obvious to combine the teachings of Hausman, Kodra, and Iwase for the purpose of efficiency.

47. Regarding claim 34, see the above rejections of claims 8, 27, and 33. The combination of Hausman, Kodra, and Iwase read on this limitation.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Daniel R. Sellers whose telephone number is 703-605-

4300. The examiner can normally be reached on Monday to Friday between 9am and 5:30pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Forester W. Isen can be reached on 703-305-4386. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

DRS

FORESTER W. ISEN
SUPERVISORY PATENT EXAMINER